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A TMS320C6416 DSP Real Time Implementation of Continuous Wavelet Filtering Module

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Abstract

This paper expounds the real time realization cochlear strategy based on wavelet transform (WT). This electrical stimulation of the brain is the explanation of voice. Electrical stimulation to real-time optimization strategy, has taken measures. These steps include the use of dynamic link libraries, make effective memory allocation and show the fixed-point algorithm. Use these steps, we have been able to reduce the processing time is about 10.49 u s in the 720 MHZ TMS320C6416 DSP board. So, we can pay attention to the execution time accounted for only 21% of the minimum two-way stimulate pulse width (25 u s/phase).

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Keywords:DSP,real time,wavelet filtering

1. INTRODUCTION

Like cochlear implants repair is used to restore hearing deaf or those with deeply of nerve deafness [1]. It provides an electrical signal, through a series of cochlear implant electrodes placed in them. It USES a phonetic processor put a microphone sound generated signals [2].

The main function of the phonetic processor input signal is decomposed into its frequency components, in the same way, analyzes the cochlea input signal in frequency healthy ingredients. Many successful cochlear implants can be attributed to the development of advanced real-time signal processing algorithm performs in the speech processor. The algorithm to a certain extent is designed to simulate the function health cochlear [2], [9].

Two important factors need to be considered when realize real-time signal processing algorithm in the application of the cochlea. First, the frequency decomposition, can do a filter USES or bank or fast Fourier transform (FFT), need to yield enough similar frequency resolution of ears. Health Fine spectral resolution is preferred transmission clues (" fine spectrum fine structure ") healing recipients. Fine resolution also need of the implementation of the key to or current stimulus current strategy. The second, and perhaps most important, happen need small signal delay [8]. This factor can consider different aspects,

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such as speed, reliability, energy consumption and functional units [5].

In this work, we explore TMS320C6416 DSP realize real-time continuous wavelet filtering strategy modules based cochlear processor. Guarantee, the time you will be enough processing components in real time, we describe the execution of this algorithm and optimized assembly code.

2. COCHLEAR IMPLANT STRATEGY FORMALISM

2.1. Stimulation Strategy Principle

The signal processor in the main functions including cochlear implants an input speech signal is divided into the many a band (12-22) in order to extract signal strength in each band for exciting implanted electrodes accordingly. In other words, the signal processor can imitate the operation of the inner ear. The most common tactic used for commercial cochlear implants continuous sampling (CIS) and alternate advanced combination code (ACE). These strategies can be used to realize the bank is a filtering method [3]. When using the method of filter Banks, a group of band pass filters used for the number of signal frequency band or channels. And then, full wave output filter rectify and through the low-pass filter extraction channel based on the envelope, pulse of electrode (figure 1).

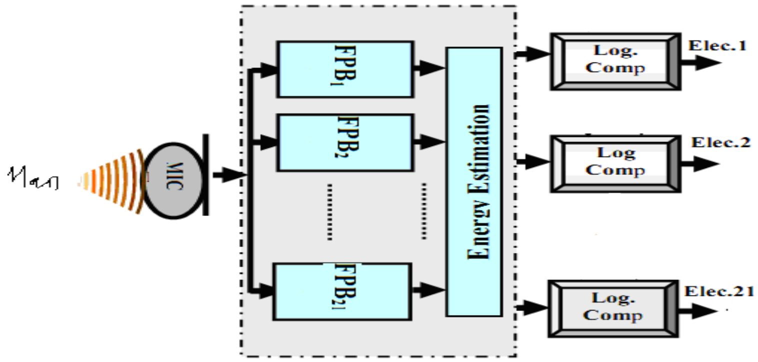


Figure 1. Bloc architecture of the proposed stimulation strategy

Wavelet method, similar to the perception of the auditory system, used in the process of speech signal input down into many a band. Based on the human auditory system, we will consider filter spectrum is divided into bank according to the critical band (table1 ERB). 16-kHz sampling frequency, we use the band-pass filter modules corresponding 18 filters which 18 energy [5].

TABLE 1. ERB CRITICAL BANDS

N	Center Freq. (f) [Hz]	inferior limit [Hz]	superior limit [Hz]
1	50	20	100
2	150	100	200
3	250	200	300
4	350	300	400
5	450	400	510
6	570	510	630
7	700	630	770
8	840	770	920

9	1000	920	1080
10	1170	1080	1270
11	1370	1270	1480
12	1600	1480	1720
13	1850	1720	2000
14	2150	2000	2320
15	2500	2320	2700
16	2900	2700	3150
17	3400	3150	3700
18	4000	3700	4400

2.2. Continuous Wavelet Transform

The continuous wavelet transform CWT is used to design a digital filter banc for a speech processing strategies used for the external part of cochlear prosthesis system [13]. The time-domain definition of CWT of f(t) is given by equation .

This analysis uses a family of functions ψ a,b(t) based on a mother wavelet .Where 'a' is a dilatation parameter, 'b' is translation parameter and $\tilde{\psi}$ (t) is the envelope of $\psi(t)$ Because of its center frequency easy choice and wavelet index form, Morlet is chosen as the mother wavelet decomposition of sound spectrum cochlear bank filter.Basically, the idea of hearing model wavelet transform is let time according to the target of the envelope mother time-varying signal features into basic levels road. This last can adjust follow the analysis model of auditory system .

The first sigma is just the scaling factor to insure the energy is the same for every mother wavelet and the second sigma adjusts the envelope of the mother wavelet $\psi(t)$ without adjusting its center frequency.

In order to share considering spectrum and fixed width of each band voice corresponding ERB model, and then puts forward the interesting research [5], finish the last expression model, give hearing mother wavelet.

3. REAL-TIME DSP IMPLEMENTATION

The main purpose of this work is to deal with speech signal real-time information and deliver the stimulation implant. Therefore, a specific platform is used to make the experimenter implementation process run time. The platform is composed of two parts: a PC and a TMS320C6416 digital signal processor (DSP) board.

3.1. TMS320C6416 DSP Board Overview

In this project, we use TMS320C6416 digital signal processor (DSP) board. C6416 recent version of the TMS320 DSPs family provide instrument. Texas VelociTI, high performance and use it very long instruction-word (VLIW) of architecture. This C6416 register file by the other support extending the eight types and 64 packaging fixed point of data types. There are two general registration files, A and B, total 32 registers [10].

3.2. Optimization Techniques

Our initial C6416 implementation is based on C language and linear assembly code. In the original version, we achieved an execution time 30 micro-pound to the necessary processing [4].

Ensure adequate time processing components in reality, we in the execution of this algorithm is described and hand-traded assembly code. The main technology is used to optimize the algorithm are explained in our side, continue in figure 2.

3.2.1. Circular addressing

C6416 processor support "circular addressing", making a piece of data processor access sequence, then automatically it wrapped in [5] the starting address of the model, it is used to access the filter transform coefficient. Through the circular addressing, memory addressing cycle changes its starting point [7].

In fact, in real-time processing, because each sample input specified level placed in the "N" buffer size (see chart 2), one is had to start reading the input buffer, in the sample position level. Here, the problem of linear addressing. We can avoid the circulation characteristics of addressing in our C6416 assembly code.

3.2.2. Register Files Partition

Know C6416 in behind, there are two general register file data access (group A and B group), each file contains 32 registers (32) A0-A31 files and B0-B31 file B). In the temporary device can be used for data, data address pointer, or status register [10]. Two of the machining instructions happened in each data access (group A and B group). So, we can optimize the time costs 50% of total time processing.

3.2.3. Fast Multipliers

The CWT filter is mathematically based on $\sum X * Psi$, where X is a vector of input data, and Psi is a vector of filter coefficients. For each “tap” of the filter, a data sample is multiplied by a filter coefficient, with the result added to a running sum for all of the taps [4].

Therefore, the principal component transform a filter algorithm is point products: take, add, multiply, add. The operation of the continuous wavelet transform filter algorithm. In fact, multiplication (usually associated with the accumulation of the product) is the most common operation signal processing [6].

So, in the algorithm every two relevant buffer (input and the transfer function), a 32-bit fixed-point data can be loaded by the same package, and through to the pace of multiplication. In this step, the plot, DOTP2 signed 16 (pack) [7] returns the dot product-between two pairs of instructions signed to 16 value, is packed with a important role. In fact, this product is the half-low load data of two is added to the product is half-and accumulation of relevant [10] destination. This process is repeated, A succession of time (A) and (B) two channels to achieve register, including the latest packets.

3.2.4. Enhanced Loading Data

DSP processors architect want to improve performance gains provide more than a clock speed and modest hardware development must find a way to get significantly more useful DSP work every day clock cycle. One way is to extend the construction of the traditional DSP concurrent execution increased unit, usually a second amplifier and adding machines.

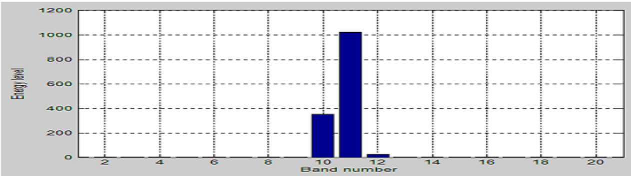
Know C6416 registers document distribution support 64-bit data types of packaging, dot the use of input data loading double word (LDNDW non-aligned) help us optimize our algorithm. In this respect, we made a accelerated four times in terms of the total number in each path of instructions executed (a) and (B) [10]. Corresponding growth, this-to conventional DSP processors can perform significantly much work in one clock cycle. For example: two DOTP2 each cycle rather than a in each path, so every four DOTP2 cycle.

4. RESULTS AND DISCUSSIONS

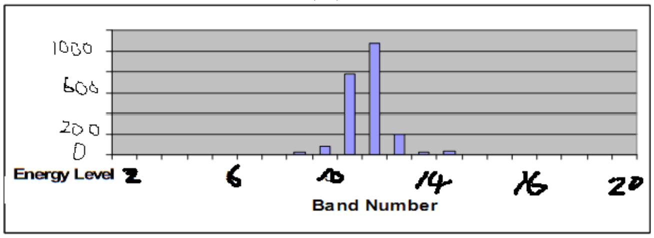
4.1. Parameterization Results

To validate the implementation of our hand-optimized algorithm, we choose to generate some harmonic signals with Matlab that makes it possible to activate and test not only each electrode alone but also real speech signals extracted from TIMIT database.

Figure 3 represents the parameterization result of a harmonic at 1.3 kHz with (m) Matlab implementation, (n) hand-optimized based TMSC6416 implementation.



(m)



(n)

Matlab results, this hand-the experimental result of the algorithm is traded is satisfactory. The cochlear filter makes it possible to find a bank in its corresponding harmonic frequency. Usually, the harmonic energy is to be located in the corresponding frequency. Minor differences are due to fixed side of the input and output in C6416 implementation data necessary for foundation (fixed point).

In addition, some experiments show that, all kinds of phonemes keep their frequency analysis of position. For example, in figure 5, we note that, "aa" guard the maximum of 8 phoneme band two types: Matlab a final version of the TMSC6416 implementation and execution.

4.2. Processing Times

In order to prove that use of each CWT optimization based, we calculated the steps necessary computing time related TMSC6416 simulation results are carried out. The compiler optimization can reduce the processing time from 47.28 to 40.60 u SEC only through the use of built-in optimization design, provide-o1, o2 and o0- -o3 compiler options.

To sum up, designed to convert the following properties: in hand-based C6416 traded assembly requirements implementing the total processing time codec 10.49 u SEC (this includes all the output). This enhancement is equal to 77.81% of the original DSP processing make a better processing performance.

5. CONCLUSIONS

In this paper, we are interested in the cochlea research, is actually a developers in different disciplines restructuring.

Our goal is to realize real-time realization model based on DSP hearing strategy use continuous wavelet transform speech processing. We put forward different optimization measures can be used to optimize the any signal processing algorithm real-time executive DSPs. In order to verify the algorithm hand code-our strategy traded in real time, we have the harmonics signal, Matlab voice signal from the extraction of some real TIMIT database.

The current implementation meets the requirements of real-time processing. The consumption of time is 10.49 u s optimization assembly code, then 16 KHZ sampling frequency. Therefore, only 21% DSP processing time of occupation.

Acknowledgement

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